Amendments to the Claims

This listing of claims will replace all prior versions, and listings, of claims in the application.

1. (Currently amended) A portable voice over Internet Protocol (VoIP) test device for testing a VoIP network, comprising:

a user interface comprising an audio input device and an audio output device;

a transceiver configured to communicate with the VoIP network;

a memory storing a test algorithm;

a codec configured to use a plurality of compression protocols;

a media access controller (MAC); and

a processor in communication with said user interface, said transceiver, said codec, said media access controller, and said memory and configured to execute said test algorithm to cause said transceiver to communicate with the VoIP network to test at least one of the group: jitter, packet loss, and latency of the VoIP network.

2. (Previously presented) The VoIP test device of claim 1, further comprising a digital signal processor in communication with said processor.

3. (Previously presented) The VoIP test device of claim 2, wherein said digital signal processor forms said codec.

4. (Currently amended) The VoIP test device of claim 1, wherein said codec ~~uses~~ is configured to use at least one of the following compression protocols: G.711a-law, G711µ-law, G.720, G.723.1, G.726, G.728, G.729, G.729A, and G.729AB2.

5. (Currently amended) The VoIP test device of claim 1, wherein said transceiver comprises a power line modem for communication with a power line communication network and communicatively coupled to a multi-prong plug configured to be plugged into an electric socket.

6. (Previously presented) The VoIP test device of claim 1, wherein said transceiver comprises an Ethernet transceiver.

7. (Currently amended) The VoIP test device of claim 1, wherein said transceiver comprises a cable modem configured to communicate information with a protocol substantially compliant with a Data Over Cable Service Interface Specification (DOCSIS) specification.

8. (Currently amended) The VoIP test device of claim 1, wherein said ~~user interface device comprises an~~ audio input device comprises a microphone and ~~an~~ said audio output device comprises a speaker.

9. (Original) The VoIP test device of claim 1, wherein said transceiver comprises a digital subscriber line (DSL) modem.

10. (Original) The VoIP test device of claim 1, wherein said user interface comprises a manual input device and a display.

11. (Previously presented) The VoIP test device of claim 1, wherein said processor is configured to execute said test algorithm to cause said transceiver to communicate with the VoIP network to test at least two of the group: jitter, packet loss, and latency of the VoIP network.

12. (Original) The VoIP test device of claim 10, further comprising a dual tone multi-frequency encoder in communication with said manual input device.

13. (Previously presented) The VoIP test device of claim 1, further comprising a communication interface port in communication with said processor.

14. (Previously presented) The VoIP test device of claim 13, wherein said communication interface port comprises a RJ-11 connector.

15. (Original) The VoIP test device of claim 13, wherein said communication interface port comprises a tip/ring interface.

16. (Currently amended) The VoIP test device of claim 1, further comprising a Power over Ethernet module <u>configured to supply power to one or more elements of the device</u>.

17. (Previously presented) The VoIP test device of claim 5, wherein said media access controller forms part of said transceiver.

18. (Previously presented) The VoIP test device of claim 1, wherein said processor is configured to execute said test algorithm to cause said transceiver to communicate with the VoIP network to test each of jitter, packet loss, and latency of the VoIP network.

19. (Original) The VoIP test device of claim 5, wherein the device receives power from a power line communication network.

20. (Original) The VoIP test device of claim 1, further comprising a network status indicator.

21. (Original) The VoIP test device of claim 20, wherein said network status indicator provides a mean opinion score (MOS) output.

22. (Previously presented) The VoIP test device of claim 1, wherein the device includes a handset and a base and said processor is disposed in said handset.

23. (Original) The VoIP test device of claim 1, wherein said processor is programmed to test the VoIP network based on at least one of the following: E-Model, Perceptual Analysis Measurement System, Perceptual Evaluation of Speech Quality, Perceptual Speech Quality Measurement (PSQM), and PSQM+.

24. (Original) The VoIP test device of claim 1, wherein said memory includes an Internet Protocol (IP) address stored therein.

25. (Previously presented) The VoIP test device of claim 1, wherein said memory includes an algorithm for requesting an IP address stored therein.

26. (Original) The VoIP test device of claim 1, wherein said memory includes a MAC address stored therein.

27. Canceled.

28. (Currently amended) A method of using a portable test device to test a VoIP network, comprising:

transmitting test signals over the VoIP network;

receiving response signals in response to transmitting said test signals;

wherein the response signals are received from the VoIP network via a codec and a media access controller <u>forming part of the device</u>;

<u>wherein said codec is configured to process signals via a plurality of compression protocols;</u>

processing said response signals to determine at least one of the group: jitter, packet loss, and latency of the VoIP network; ~~and~~

presenting an indication of a result of said processing of ~~the VoIP network~~ <u>said response signals</u> to the user<u>; and</u>

<u>providing an audio output device and an audio input device in the test device to facilitate bi-directional VoIP communications over a VoIP network by the user</u>.

29. (Original) The method of claim 28, wherein the processing comprises at least one of time-frequency mapping, frequency warping, intensity warping, loudness scaling, asymmetric masking, and cognitive modeling.

30. (Previously presented) The method of claim 28, wherein said presenting an indication comprises indicating at least one of the following: incorrect Internet Protocol configuration, incorrect gateway address designation, signal echo, and call drop out.

31. (Original) The method of claim 28, further comprising determining whether the VoIP network is operable to communicate voice data according to predetermined voice communication parameters.

32. (Previously presented) The method of claim 28, wherein said processing comprises determining signal distortion.

33. (Previously presented) The method of claim 28, wherein said processing comprises determining each of, signal delay, jitter, and packet loss of the VoIP network.

34. (Previously presented) The method of claim 28, wherein said processing comprises determining at least two of the group: packet jitter, packet loss, and latency of the VoIP network.

35. (Original) The method of claim 28, wherein said indication comprises a MOS indication.

36-48 Canceled.

49. (Currently amended) A method of ~~testing~~ <u>using a test device to test</u> a VoIP network, comprising:

receiving an input from a user interface;

executing a test algorithm;

transmitting a first test signal over the VoIP network;

receiving a second signal from the VoIP network;

processing the received second signal via a codec and a media access controller; ~~and~~

wherein said codec is configured to use a plurality of compression protocols;

processing said second signal to determine a jitter, a packet loss, and a latency of the VoIP network; and

providing an audio output device and an audio input device in the test device to facilitate bi-directional VoIP communications over a VoIP network by a user.

50-55. Canceled.

56. (New) The method of claim 49, wherein said codec is configured to use at least one of the following compression protocols: G.711a-law, G711μ-law, G.720, G.723.1, G.726, G.728, G.729, G.729A, and G.729AB2.

57. (New) The method of claim 49, wherein the network comprises a cable network.

58. (New) The method of claim 49, wherein the audio input device comprises a microphone and the audio output device comprises a speaker.

59. (New) The method of claim 49, wherein the VoIP network comprises a digital subscriber line (DSL) network.

60. (New) The method of claim 49, wherein said processing comprises processing said second signal based on at least one of the following: E-Model, Perceptual Analysis Measurement System, Perceptual Evaluation of Speech Quality, Perceptual Speech Quality Measurement (PSQM), and PSQM+.

61. (New) The method of claim 60, wherein the user interface is communicatively coupled to a dual tone multi-frequency encoder.

62. (New) The method of claim 49, further comprising providing power via a Power over Ethernet module.

63. (New) The method of claim 49, wherein the VoIP network comprises a power line communication network.

64. (New) The method of claim 63, wherein the device receives power from the power line communication network.

65. (New) The method of claim 49, wherein the VoIP network comprises an Ethernet network.